Note For robbed-bit interfaces, if the `default-call-type` is `voip` in the `t1` profile, then the bearer capability in the setup message is altered to indicate that this is "packetized voice."

In-bound calls received on all channels of this T1 will be processed by the TAOS unit as VoIP calls.

Enabling DTMF R2 signaling for E1 lines

MultiVoice Gateway can process Dual Tone Multi-Frequency (DTMF) tones over R2 signaling trunks to provide support for processing either country-specific R2 signaling (MFC-R2) or DTMF signaling over trunks that support standard R2 signaling.

MultiVoice Gateways can support DTMF R2 signaling generated by smaller European network switches and PBXs. MultiVoice implements DTMF tone processing using the R2 signaling standard defined by the International Telecommunications Union Telecommunication sector standard (ITU-T) Q.400, *Specifications of Signaling System R2 Definition and Function of Signals -- Forward Line Signals.*

A channelized E1 slot card uses one of the following channelized associated signaling (CAS) types:

- R1
- R2 or any R2 variant
- DTMF-R2

Note Only one signaling type can be used on an TAOS unit channelized E1 slot card.

To support DTMF-R2 detection, MultiVoice requires the following:

- Connection to E1 trunks that are attached to a switch that supports the ITU-T R2 signaling standard.
- The switch must generate and/or relay the high-frequency/low-frequency tone combinations generated by normal touchtone dialing to the MultiVoice Gateway.
- R2 signaling must be enabled on the MultiVoice Gateway. Verify that the R2 signaling parameter is enabled—check the base profile for `r2-signaling-enabled`=yes.

Detection of DTMF R2 signals is enabled from the `e1` profile.

DTMF tone detection

When processing tones for DTMF R2 signaling, the MultiVoice Gateway performs as follows:

- Upon detection of an inbound call, allocate a DSP for detecting DTMF tones, capturing DTMF digits as they are received from the switch.
- Upon receipt of an outbound call (from the packet network), allocates a DSP for generating DTMF tones, sending the first DTMF tone for 70ms, followed by 70ms of silence. This tone-silence sequence is repeated until all digits are sent to the telephone switch.

Note A DSP can successfully generate and detect test tones for E1 SS7 continuity testing without impacting detection and processing of E1-R2 signaling. Specifically, the required frequency for the E1 SS7 continuity check, 1780+-20Hz and 2000+-20Hz with the sending tone level of -12+-1dbm0, as defined in ITU

Telecommunication sector standard (ITU-T) Q.724, *Specifications of Signalling System No. 7 - Telephone user part* (1988), International Telecommunications Union, are detected by the DSP. The DSP differentiates between tones in this same frequency ranges which are used for E1-R2 signaling and the E1 SS7 continuity check.

The following is an example of how to enable DTMF R2 signaling on an APX or MAX TNT E1 line slot card.

```
admin> read e1 { 1 1 7 }
E1/{ 1 1 7 } read
admin> set signaling-mode=dtmf-r2-signaling
admin> set collect-incoming-digits=no
admin> set e1-inter-digit-timeout=3000
admin> write
E1/{ 1 1 7 } written
```

Using Signaling-Mode parameter for DTMF R2 signaling on E1 lines

One of the settings for the `signaling-mode` parameter in the `e1` profile enables DTMF R2 signaling detection and processing in the `e1` line profile.Setting the value of the `signaling-mode` parameter to `dtmf-r2-signaling` value enables the TAOS unit to recognize and respond to the DTMF R2 signal set during voice and data calls. Once selected, DTMF R2 detection is enabled with the next VoIP call.

The following dependencies apply when `signaling-mode=dtmf-r2-signaling`:

- `collect-incoming-digits` must be enabled (`collect-incoming-digits=yes`).
- Assigning a lower value (such as 600 to 3000) to the `e1-inter-digit-timeout` parameter improves call setup times. Assigning a higher value (such as 3001 to 6000) improves detection of DTMF.
- DTMF R2 detection is only supported when R2 signal processing is enabled for the TAOS unit. The Base profile should contain the following setting:

  `r2-signaling-enabled=yes`

Enabling and debugging Feature Group D signaling support for T1 lines

MultiVoice supports a subset of the Telecordia requirements for Feature Group D (FGD) signaling for Voice over IP processing, such as passing Automatic Number Identification II (II) information, Calling-Party-Number and Called-Party-Number as MFR1 tones on inc-wink signaled trunks. A MultiVoice Gateway can manage interworking between Access Tandem carriers and traditional toll service carriers for VoIP calls. It also provides basic support for trunk-side access with Equal Access dialing capability, pre subscription, and enhanced signaling options for Automatic Number Identification as specified by Requirement GR-690-CORE, *Exchange Access Interconnection FAS 20-24-0000* (Oct. 1995), Telecordia Systems (formerly Bellcore).

Feature Group D access service with equal access multifrequency signaling is characterized by two-stage outpulsing when connection is made through the access tandem. The first stage provides information to the AT for selection of a carrier and the route to take to that carrier. The second stage provides the carrier with both the calling-party-number (and, optionally, ANI) and the Called-Party Number (address or destination number). Overlap outpulsing is used to transmit this information using multifrequency signaling.

Pass-through of equal access signaling can be enabled for T1 inband trunks from the T1 line profile. When FGD signaling is enabled, MultiVoice Gateways can recognize and process the single-dialed access carrier destination, (such as: 1,2025551212 or 1,1010220,2025551212). To support access carrier billing, a MultiVoice Gateway passes the Calling-Party-Number, ANI information digits and Called-Party-Number. The ANI information digits, a two-digit code, classifies the Calling-Party-Number by tariff type (such as coin, 800 service, or POTs).

MultiVoice also manages interworking when connecting VoIP calls between Access Tandem networks and traditional toll service networks. An egress MultiVoice Gateway can be configured to receive Calling-Party Number, ANI information digits, and Called Party-Number from an Access Tandem switch and connect that call to a switched telephone network that supports Feature Group C (FGC), that is, traditional toll service and switching. FGC includes automatic number identification of the calling party, answerback, and disconnection supervision. FGC service predates the breakup of the Bell System.

T1 profile

Feature Group D licensed software for MultiVoice Gateways adds FGD signaling options to the `signaling-mode` parameter in the `t1` profile to enable inband signal processing of FGD signals, and interworking of MultiVoice networks between Access Tandem carriers and traditional toll-service carriers.

fgd-signaling-enabled parameter

Following installation of TAOS, each MultiVoice Gateway must be loaded with licensed software code to enable processing of Feature Group D signaling. When enabled, `fgd-signaling-enabled` parameter appears in the base profile.

During manufacturing or software upgrade of MultiVoice Gateways, the installation binaries used to install TAOS on the TAOS unit asks if FGD support should be enabled. FGD support can only be enabled or disabled by reinstalling the licensed software on the TAOS unit.

When support for FGD is enabled, the `fgd-signaling-enabled` parameter is added to the base profile, as illustrated:

```
admin> read base
BASE read (read-only)

admin> list
[in BASE]

shelf-number = 1
software-version = 8
software-revision = 0
software-level = ""
manufacturer = dba-ascend-mfg
....
fgd-signaling-enabled = yes
```

Using the signaling-mode parameter to configure FGD signaling

The `signaling-mode` parameter in the `t1:line-interface` subprofile identifies the type of call signal received from the ingress switched telephone network and the type of call signals passed to the egress switched telephone network. New values may be assigned to the Signaling-Mode which enable processing of FGD inband signaling for connecting Equal Access calls, and support interworking between Access Tandem and traditional toll service networks.

The `signaling-mode` parameter may be assigned the following values to enable FGD signaling support:

Parameter	Setting
`inband-fgd-in-fgd-out`	Configures the MultiVoice Gateway to expect to receive call signaling data in FGD format, and connect VoIP calls to the egress switched telephone network, sending call signaling data in FGD format.
`inband-fgd-in-fgc-out`	Configures the MultiVoice Gateway to expect to receive call signaling data in FGD format, and connect VoIP calls to the egress switched telephone network sending call signaling data in FGC format.
`inband-fgc-in-fgc-out`	Configures the MultiVoice Gateway to expect to receive call signaling data in FGC format, and connect VoIP calls to the egress switched telephone network sending call signaling data in FGC format.
`inband-fgc-in-fgd-out`	Configures the MultiVoice Gateway to expect to receive call signaling data in FGC format, and connect VoIP calls to the egress switched telephone network sending call signaling data in FGD format.

Changes made to the `signaling-mode` parameter take effect with the next VoIP call. The following example illustrates how to enable a MultiVoice Gateway to receive call signaling data in FGD format and send call signaling data to the egress switched telephone network in FGC format signalling using the `signaling-mode` parameter.

```
admin> read t1 { 1 1 1 }
T1/{ 1 1 1 } read

admin> list line
[in T1/{ shelf-1 slot-1 1 }:line-interface]

enabled = yes
frame-type = esf
encoding = b8zs
signaling-mode = inband
.........
ss7-continuity = { loopback single-tone-2010 }

admin> set signaling-mode=inband-fgd-in-fgc-out

admin> write
T1/{ 1 1 1 } written
```

For the `signaling-mode` parameter to include FGD signaling options, the MultiVoice Gateway must have licensed software code for FGD processing.

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Feature Group D signaling timing

To ensure wideranging interoperability with available access tandem switches, MultiVoice uses the middlerange Feature Group D Signaling Timing.

- Wait up to 210 msecs for first wink from the Access Carrier. Requirement GR-690-CORE specifies a range of 140-290 msecs.
- Wait up to 5 seconds to receive the first digit. After sending the first wink on receipt of an off hook, the MultiVoice Gateways waits for 5 seconds before reporting a time-out error if the first digit signal is not received.
- Wait up to 4 seconds for a wink from the Access Carrier. Requirement GR-690-CORE specifies Access Tandem switches wait for up to 4 seconds for this signal.

Debugging Feature Group D signaling

To collect debugging information for Feature Group D inband signal processing on a TAOS unit, enter the following commands:

```
TnT01> open 1 1          (where the T1 slot card is)
TnT01> fgdtoggle         (turn on fgd signalling debugging)
TnT01> debug on
```

The debug command displays information similar to the following for Feature Group D signal processing.

```
----------------------------------------------------------------------
0 Connections, 0 Sessions      | TNTFGD Status
                               | Serial number: 9340872 Version: 9.0a0e0
                               |
                               |  Rx Pkt:     1057139
                               |  Tx Pkt:      163995
                               |     Col:        2244
                               |
                               |
                               | 02/29/2000 17:50:27  Up: 7 days, 00:17:30
                               | "T3 Tru+ 1/03/00 LA la la la la la la la
                               | "T3 Fiv+ 1/03/01 LA F------- ------- -------
                               | "T3 CDX" 1/03/02 LA F-------- -------- -----
                               |                     ^
                               |                     |
                               |                     +-- display (F)for FGD line
                               |
                               |
-----------------------------------
```

Enabling collections of variable length dial strings without EOP

The MultiVoice Gateway collects variable-length dial strings without using end-of-pulse (EOP) signaling. In certain areas outside the continental United States where E1 MFC-R2 signaling is used for switched network operations, the length of E.164 addresses vary. End-of-pulse (EOP) detection is not efficient, since the network may be unable to complete the call as a result of network conditions.

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Collection of up to 15-digits

MultiVoice Gateways are compatible for use on European telephone systems that use E.164 addresses that are up to 15 digits long, without waiting for an end-of-pulse signal.

Previously, MultiVoice Gateways could be configured to collect dial strings of up to only 11 digits. For European networks using dial strings that were 12 digits or longer, a MultiVoice Gateway could only be configured to wait for the end-of-pulse signal to confirm it received all the dialed digits.

The number of configurable digits to 15 for the E1 line `number-complete` parameter is set in the `e1:line-interface` sub-profile.

The `number-complete` parameter enables detection and collection of up to 15 digits for inbound dialed telephone numbers on MultiVoice Gateways using E1 trunks supporting inband CMF R2.

The parameter now accepts values from `0-digits` through `15-digits`, `end-of-pulse`, and `time-out` as valid entries.

The following example illustrates how to enable the collection of 15 digit dial strings on a TAOS unit:

```
admin> read e1 { 1 1 7 }
E1/{ 1 1 7 } read

admin> set number-complete=15-digits

admin> write
E1/{ 1 1 7 } written
```

The following dependencies apply to this parameter:

- The `number-complete` parameter defaults to `N/A` when the `signaling-mode` parameter is assigned any of the following values:
  - `e1-kuwait-signaling`
  - `isdn`
  - `p7`
  - `dpnss`
  - `none`

Time-out processing

A MultiVoice Gateway can be configured to time-out, followed by pulse signals, as specified by ITU-T Recommendation Q.442, *Specifications of Signalling (sic) System R2 interregister Signalling (sic), Pulse Transmission of Backward Signals A-3, A-4, A-6 or A-15* (1993), International Telecommunications Union.

Using time-out allows a MultiVoice Gateway to delay processing of a dialed number string, even after receiving the last digit, to allowing the resources on the switched network additional time to become available, before continuing with call processing.

To implement this feature, modify entries for the `number-complete` and `inter-digit-time-out` parameters.

The `number-complete` parameter sets the condition the MultiVoice Gateway uses to determine the length of the dial string. For E1 MFC-R2, the MultiVoice Gateway

continues to collect digits until the on/off pulsing using to transmits the dial string is complete.

The `number-complete` parameter now accepts the following value:

Parameter value	Description
`time-out`	Configures the MultiVoice Gateway to reset the network idle timer after the initial digit is received, and then wait for silence. Once silence is detected, wait the interval specified by the `inter-digit-time-out` parameter for next digit. The MultiVoice Gateway continues to collect digits, while waiting for the network idle timer to expire before continuing with call processing.

The following illustrates how to configure a MultiVoice Gateway to determine the length of a dial string using time-out processing.

```
admin> read e1 { 1 1 1 }
E1/{ 1 1 1 } read

admin> list line
[in E1/{ shelf-1 slot-1 1 }:line-interface]

enabled = yes
frame-type = esf
encoding = b8zs
signaling-mode = inband
.........
ss7-continuity = { loopback single-tone-2010 }

admin> set number-complete=time-out

admin> write
```

E1 MFC-R2 signaling is country specific. The `signaling-mode` parameter, and the `country` parameter in the `system` profile, must be set for the country-appropriate signaling in order for the MultiVoice Gateway to properly detect dialed digits.

In the `e1:line-interface` subprofile, the `inter-digit-time-out` parameter controls how long a MultiVoice Gateway will wait after receiving the last digit of a dial string before declaring DNIS/ANI collection complete. When using inband signaling (T1, MF R2), a TAOS unit waits until this interval has elapsed to ensure it has received all audible tones used to transmit DNIS/ANI across the trunk.

The `inter-digit-time-out` parameter accepts values between 100 and 6000msec. This parameter defaults to 3000msec (3 seconds). For configurations supporting E1 MRC R2 signaling, the `inter-digit-time-out` parameter accepts values between 200 and 6000msec.

The following illustrates how to configure the interdigit timer on a MultiVoice Gateway to wait one second (1000msec) in between dialed digits before continuing with call processing.

```
admin> read e1 { 1 1 1 }
E1/{ 1 1 1 } read
```

```
admin> list line
[in E1/{ shelf-1 slot-1 1 }:line-interface]
enabled = yes
frame-type = esf
encoding = b8zs
signaling-mode = inband
.........
ss7-continuity = { loopback single-tone-2010 }
admin> set inter-digit-time-out=1000
admin> write
```

E1 MFC-R2 signaling is country specific. The `signaling-mode` parameter in the `e1` profile and the country parameter in the `system` profile must be set for the country-appropriate signaling for the MultiVoice Gateway to properly detect dialed digits.

Processing ANI and DNIS for H.323 VoIP

Regardless of whether a TAOS unit is configured for single stage-dialing or two-stage dialing, the master DSP or slave DSP parses the dual-tone multifrequency (DTMF) tones in the following order, based on how the tones were processed by the PSTN/ switch:

CLID*DNIS
DNIS
CLID*DNIS*

When the local TAOS unit connects with the destination TAOS unit involved in a call, the reported DNIS and ANI/CLID, which are included as part of the call setup message(s), are used as follows:

String	Description
ANI/CLID	The Automatic Number Identifier or Calling Number IDentification string of the caller's telephone: • At the destination gateway, this number identifies the origin of the call. Once received by the destination gateway, the number is exported for use by gateway applications, passed back to the calling gateway to confirm call setup, etc. • At the local gateway, this number is collected then forwarded to the destination gateway and the MultiVoice Access Manger, where it may be used for authentication, call reporting, third-party billing applications, etc. • At the destination gateway, this number is passed to the PSTN and used for initiating local switched network services that process CLID (such as Caller ID, call waiting, last number redial, etc.).

String	Description
DNIS	The Dialed Number Identification Service string that identifies the called telephone: • At the destination gateway, this number is the destination telephone number dialed by the MultiVoice Gateway. • At the local gateway, this number is the dial string entered by the caller for the destination telephone number.

Configuring 480 ports for G.711-encoded VoIP-only calls

The MultiDSP 288-port slot card (APX-SL-DSP-3) in an APX 8000 or APX 1000 supports the following modes:

- 288 ports of universal port traffic (that is, simultaneous V.110 and V.92 modem, High-Level Data Link Control (HDLC), Personal Handyphone System (PHS), T.38 Fax, VoIP with G.711, G.729(A) audio codecs). This is the default mode.
- 480 ports of G.711 VoIP-only calls for H.323 and IP device control (IPDC) protocols.

Sites that use the G.711 codec in their networks have the option to configure 480 ports, which lowers the cost of network equipment and operation.

To support 480 ports of G.711 VoIP-only traffic, the following conditions must be met:

- The universal gateway must be an APX 8000 or APX 1000 unit.
- The MultiVoice software license for each universal gateway must be enabled. Additional software licenses or hardware are not required.
- The `frames-per-packet` parameter in the `voip` profile must be set to 4 or greater. The maximum number of frames per packet you can set is 10. If a number lower than 4 is set, the value is automatically reset to 4 and a log message is generated.

Note The value specified for the `frames-per-packet` parameter (4 - 10) must be supported by all MultiDSP slot cards in the chassis. If using the IPDC protocol, the softswitch should use four frames per packet for all calls.

Transparent fax and modem

Regular modem (for example, v.90/v.92) and T.38 fax capability are not supported in the 480 port configuration. However, when configured for 480 ports, the slot card uses two frames per packet for transparent fax and modem.

The slot card can handle only a maximum of 96 transparent fax and modem calls per quadrant. When using priority-based call routing (see "Priority-based call routing" on page 2-25), all G.711 calls are routed to the slot card. Even if other MultiDSP slot cards also handle transparent fax, Lucent recommends allowing only a maximum of 96 transparent calls per quadrant.

Compatibility with other MultiDSP slot cards

When using IPDC and the slot card is configured for 480 ports of G.711 VoIP-only traffic, the MultiDSP 288-port slot card can coexist with 48-port and 96-port MultiDSP universal port slot cards that use other codecs.

However, when using H.323 and the slot card is configured for 480 ports of G.711 VoIP-only traffic, only G.711 is supported by all MultiDSP universal port slot cards.

New value for subtype parameter

When the slot card is activated initially, the system checks for the presence of a madd-slot-config profile.

If the profile exists, the system examines the subtype parameter to determine the number of ports the slot card supports. If the profile does not exist, then the slot card operates with the default 288 universal ports.

The values for the subtype parameter are:

Parameter Value	Description
480-voip-ports	Configures the slot card for 480 ports.
288-univ-ports	Configures the slot card for 288 ports. Default mode.

Configuring the slot card for 480 ports

To configure the MultiDSP 288-port slot card for 480 ports, create a new madd-slot-config profile and specify a value for the subtype parameter.

The following example configures the slot card that resides in shelf 1, slot 8:

```
admin> new madd-slot-config { 1 8 0 }
MADD-SLOT-CONFIG/{ shelf-1 slot-8 0 } read

admin> list
[in MADD-SLOT-CONFIG/{ shelf-1 slot-8 0 } (new)]
slot-address* = { shelf-1 slot-8 0}
subtype = 288-univ-ports

admin> set subtype = 480-voip-ports
admin> wr
```

Resetting the slot card

After making changes to the madd-slot-config profile, verify that there are no active calls being processed by the slot card. Then reset the slot card or reset the entire universal gateway. To reset a slot card located in shelf 1, slot 8, proceed as follows:

1. **Check to see if the slot card is currently processing active calls:**

```
admin> modem -i | grep "1 8"
Modems allocated/in-use:
  Modem { 1 8 3 } ( Up Assign UP UP ENABLE )
```

2. **While waiting for active calls to be discontinued, prevent new calls from being routed into this slot card by disabling the modems:**

```
admin> mdmdisable 1 8
```

3. **Keep checking the status of the current calls until all calls are no longer being processed:**

```
admin> modem -i | grep "1 8"
```

You should see the following message before resetting the slot card:

```
Modems allocated/in-use:
  Modem { 1 8 3 } ( Up Assign UP UP ENABLE )
```

4. **Bring the slot down with the following command:**

   ```
   admin> slot -d 1 8
   ```

5. **Remove the leftover profiles from the system with the following command:**

   ```
   admin> slot -r 1 8
   Slot 1/8, state change forced
   ```

6. **Activate the slot card with the following command:**

   ```
   admin> slot -u 1 8
   ```

7. **To show status of the slot card, enter the following command:**

   ```
   admin> sh
   Controller { left-controller } (PRIMARY):
                           Reqd   Oper    Slot Type
   { shelf-1 slot-1 0}     UP     UP      8t1-card
   { shelf-1 slot-2 0 }    DOWN   RESET   ether3-card
   { shelf-1 slot-7 0 }    DOWN   RESET   t3-card
   { shelf-1 slot-8 0 }    UP     UP      madd3-voip-480
   ```

Reverting to universal port mode

To revert to using 288 universal ports, use the following commands. The example assumes that the slot card has been installed in shelf 1, slot 8:

```
admin> read madd-slot-config { 1 8 0 }
MADD-SLOT-CONFIG/{ shelf-1 slot-8 0 } read

admin> list
[in MADD-SLOT-CONFIG/{ shelf-1 slot-1 0 } (new)]
slot-address* = { shelf-1 slot-8 0}
subtype = 480-voip-ports


admin> set subtype = 288-univ-ports
admin> wr
```

After making changes to the `madd-slot-config` profile, reset the slot card or reset the entire universal gateway (see “Resetting the slot card” on page 2-66 for details).

In-call DTMF detection for IPDC

You can configure your TAOS unit to allow Softswitch to direct the MultiVoice Gateway to perform in-call DTMF detection and notification while a packet call is in progress. This is accomplished by modification of the RCCP, RMCP, and NTN messages. Any DTMF digits entered during the call while DTMF detection is enabled are still played out to the other party.

Note In-call DTMF detection is supported for packet calls, but not for time-division multiplexing (TDM) calls. Also, this feature requires obtaining a pre-paid billing application.

IPDC messages that support in-call DTMF detection

Following are IPDC messages (as defined in IPDC specification *Level 3 Communications, Internet Protocol Device Control (IPDC), Revision 0.15)* that support in-call DTMF dection and notification.

Table 2-19. IPDC messages supporting in-call DTMF detection and generation

IPDC message	Tag	Tag values
RCCP	0x75 (Constant DTMF Tone Detection)	• 0x00 - DTMF tone detection off • 0x01 - DTMF tone detection on If a tag is missing, DTMF tone detection is off.
RMCP	0x75 (Constant DTMF Tone Detection)	• 0x00 - DTMF tone detection off • 0x01 - DTMF tone detection on If a tag is missing, there is no affect on the current state of detection.
NTN	0x49 (Tone Type)	• 0x01 - DTMF tone
NTN	0x33 (Tone String)	This tag contains the DTMF digit that was detected. The length of this tag value is 1.

Call Flow

In the following call flow, a packet call is setup with DTMF detection enabled. After two DTMF digits are entered, the call is modified to disable DTMF detection.

```
  TAOS Unit                        TAOS Unit
  (incoming)      Softswitch       (outgoing)
  ----------      ----------       ----------
       |                |                |
       |                |      RCCP      |
       |                |--------------->|
       |                |      ACCP      |
       |                |<---------------|
       |      RCCP      |                |
       |<---------------|                |  Tag 0x75 = 0x01
       |      ACCP      |                |
       |--------------->|                |
       |      RTP       |                |
       |<------------------------------->| Voice exchange in
   progress
```


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```
                |              |              |
user enters     |     NTN      |              |
a dtmf digit    |------------->|              | Tag 0x33 = <digit>
                |              |              |
user enters     |     NTN      |              |
a dtmf digit    |------------->|              | Tag 0x33 = <digit>
                |              |              |
                |     RMCP     |              |
                |<-------------|              | Tag 0x75 = 0x00
                |     AMCP     |              |
                |------------->|              |
user enters     |              |              |
a dtmf digit    |              |              |
                |              |              |
user enters     |              |              |
a dtmf digit    |              |              |
                |              |              |
```

Interaction with break-in voice announcements

If in-call DTMF detection is enabled and a break-in announcement is played (see "Break-in voice announcements in IPDC" on page 4-6 for details), the first DTMF entered will stop the announcement:

```
               TAOS Unit                    TAOS Unit
               (incoming)    Softswitch     (outgoing)
               ----------    ----------     ----------
                    |              |     RCCP     |
                    |              |------------->|
                    |              |     ACCP     |
                    |              |<-------------|
                    |     RCCP     |              |
                    |<-------------|              | Tag 0x75 = 0x01
                    |     ACCP     |              |
                    |------------->|              |
                    |     RTP      |              |
                    |<--------------------------->| Voice exchange in
                    |              |              | progress
                    |              |              |
                    |     STN      |              |
                    |<-------------|              |
                    |     ASTN     |              |
                    |------------->|              |Announcement starts
                    |              |              |
user enters         |     NTN      |              |
a dtmf digit        |------------->|              |Tag 0x33 = <digit>
                    |     ASTN     |              |
                    |------------->|              |Announcement stops
```

An RMCP that is received by the MultiVoice Gateway while a break-in announcement is playing is rejected. An MRJ will be sent with Tag 0xFE (Cause Code) set to 0x65 (Message Not Compatible With Call State).

Note If DTMF is being carried inband, then the first DTMF digit entered during a break-in announcement is not played out to the other party.

Call re-origination

In-call DTMF detection can be combined with existing IPDC support on the MultiVoice Gateway to provide a call re-origination application.

Using in-call DTMF detection, the MultiVoice Gateway forwards DTMF received during an active packet call to the Softswitch. The DTMF is sent in the NTN message, one digit per message. The Softswitch monitors the received DTMF stream for a pattern (for example, **9) that indicates that the calling party wishes to terminate the active call and start a new call.

The Softswitch then sends an RCR, waits for the ACR, then sends an LTN to start the two-stage dialing for the next call, while maintaining the signaling for the incoming CIC with the PSTN.

The RCR tells the incoming MultiVoice Gateway to terminate the VoIP call. This tears down the VoIP call route and frees the resources associated with the VoIP call. The ensuing LTN would be identified as the first one for a call. This tells the incoming MultiVoice Gateway to setup a new VoIP call route.

The following call flow shows how in-call DTMF detection is utilized for call re-origination.

```
SS7 Network    TAOS Unit                    TAOS Unit    SS7 Network
(incoming)     (incoming)     Softswitch    (outgoing)   (outgoing)
-----------    ----------     ----------    ----------   -----------
        |   IAM    |              |             |              |
        |------------------------>|             |              |
        |          |     LTN      |             |              |
        |          |<-------------|             |              |
        |          |     ALTN     |             |              |
        |          |------------->|             |              |
        |   ACM    |              |             |              |
        |<------------------------|             |              |
        |   ANM    |              |             |              |
        |<------------------------|             |              |
A:                 |     STN      |             |              |
        play       |<-------------|             |              |
        announcement     ASTN     |             |              |
        to enter   |------------->|             |              |
        dnis       |     ASTN     |             |              |
                   |------------->|             |              |
                   |     ALTN     |             |              |
        dnis entered------------->|             |              |
                   |              |             |     IAM      |
        |          |              |--------------------------->|
        |          |              |             |     ACM      |
        |          |              |<---------------------------|
```

```
 |                |                |                | ANM            |
 |                |                |<-------------------------------|
 |                |                |     RCCP       |                |
 |                |                |--------------->|                |
 |                |                |     ACCP       |                |
 |                |                |<---------------|                |
                  |     RCCP       |                |                |
packet call       |<---------------|                |                |
set up            |     ACCP       |                |                |
dtmf detection    |--------------->|                |                |
enabled           |     RTP        |                |                |
                  |<------------------------------->|                |
                  |                |                |                |
user enters       |     NTN        |                |                |
a dtmf digit      |--------------->|                |                |
                  |                |                |                |
user enters       |     NTN        |                |                |
a dtmf digit      |--------------->|                |                |
                  |                |                |                |
softswitch recognizes              |                |                |
the 2 digits as the                |                |                |
pattern to initiate                |                |                |
the next call                      |                |                |
                  |                |     REL        |                |
softswitch        |                |------------------------------->|
releases other    |                |     RCR        |                |
end (both         |                |--------------->|                |
packet and        |                |     ACR        |                |
circuit legs      |                |<---------------|                |
of call)          |                |     RLC        |                |
                  |                |<-------------------------------|
                  |                |                |                |
                  |     RCR        |                |                |
release           |<---------------|                |                |
packet call       |     ACR        |                |                |
                  |--------------->|                |                |
                  |                |                |                |
                  |     LTN        |                |                |
start a new       |<---------------|                |                |
2-stage call      |     ALTN       |                |                |
                  |--------------->|                |                |
(goto "A:"        |                |                |                |
 above)           |                |                |                |
```

Note Call re-origination when signaled over SS7 VoIP does not use the voip profile parameters sequential-call-enable and next-call. These parameters are used when call re-origination is signaled over H.323 VoIP.

End-of-call break-in voice announcements

This section illustrates call flows for the special case of playing a break-in voice announcement at the end of a call. Such an announcement could be played either before or after the call is released, with VoIP call persistence enabled or disabled.

- End-of-call break-in announcement is played after call release, with VoIP call persistence mode disabled.

```
 TAOS Unit                         TAOS Unit
(incoming)       Softswitch       (outgoing)
----------       ----------       ----------
        |                 |                 |
        |     RTP         |                 |
        |<--------------------------------->|
        |                 |      RCR        |
        |                 |---------------->|
        |                 |      ACR        |
        |                 |<----------------|
        |     RCR         |                 |
        |<----------------|                 |
        |     ACR         |                 |
        |---------------->|                 |
        |     STN         |                 |
        |<----------------|                 |
        |     ASTN        |                 |
        |---------------->|                 |
        |     ASTN        |                 |
        |---------------->|                 |
```

The STN results in the setup of a new VoIP call route. Note that this messaging is possible without support for break-in announcements by using existing capabilities.

- End-of-call break-in is played after call release, with VoIP call persistence mode enabled.

```
 TAOS Unit                         TAOS Unit
(incoming)       Softswitch       (outgoing)
----------       ----------       ----------
        |                 |                 |
        |     RTP         |                 |
        |<--------------------------------->|
        |                 |      RCR        |
        |                 |---------------->|
        |                 |      ACR        |
        |                 |<----------------|
        |     RCR         |                 |
        |<----------------|                 |
        |     ACR         |                 |
        |---------------->|                 |
        |     STN         |                 |
        |<----------------|                 |
        |     ASTN        |                 |
        |---------------->|                 |
        |     ASTN        |                 |
        |---------------->|                 |
        |     RCR         |                 |
        |<----------------|                 |
        |     ACR         |                 |
        |---------------->|                 |
```

The STN results in the setup of a new VoIP call route. A second RCR is required to destroy the VoIP call route setup by the STN to play the break-in announcement.

Note The exception to this is if call re-origination is in progress. In this case, the second RCR is replaced with a LTN/STN/RCCP signaling at the start of the next call. The VoIP call route that was set up for the break-in announcement is then re-used.

- End-of-call break-in is played before call release, VoIP call persistence mode enabled or disabled.

```
 TAOS Unit                          TAOS Unit
(incoming)      Softswitch         (outgoing)
----------      ----------         ----------
     |                |                 |
     |      RTP       |                 |
     |<-------------------------------->|
     |                |       RCR       |
     |                |---------------->|
     |                |       ACR       |
     |                |<----------------|
     |      STN       |                 |
     |<---------------|                 |
     |      ASTN      |                 |
     |--------------->|                 |
     |      ASTN      |                 |
     |--------------->|                 |
     |      RCR       |                 |
     |<---------------|                 |
     |      ACR       |                 |
     |--------------->|                 |
```

This messaging is possible with support for break-in announcements. The STN utilizes the existing VoIP call route for the call. This reduces gateway processing, but adds extra seconds to the call.

DTMF playout for IPDC

TAOS supports Dual Tone Multi-Frequency (DTMF) digit playout signalled by the IPDC STN message. It plays the DTMF digits utilizing a Digital Signal Processor (DSP) on the line card or a MultiDSP card. A DSP on the MultiDSP card can be associated with a VoIP call route, so Softswitch can direct a MultiVoice Gateway to play the DTMF digits during an active VoIP call.

Note Refer to *Level 3 Communications, Internet Protocol Device Control (IPDC), Revision 0.15* specification for an explanation of all messages and tags that are referred to in the following sections.

IPDC messages that support DTMF playout

Following are IPDC messages (as defined in IPDC specification *Level 3 Communications, Internet Protocol Device Control (IPDC), Revision 0.15)* that support DTMF playout.

Table 2-20. IPDC messages supporting DTMF playout

IPDC message	Tag	Tag values
STN	0x49 (Tone Type)	Specify 0x01 (DTMF tone) to play DTMF digits.
	0x4A (Apply/Cancel Tone)	The following value is supported: 0x00 (Apply tone)
	0x32 (Num Tones)	The value associated with this tag indicates the number of DTMF digits to be played out.
	0x33 (Tone String)	The value associated with this tag contains the DTMF digits to be played out.
ASTN	0x36 (Completion Status)	The following values are returned: • 0x00 (Operation successful) • 0x01 (Operation failed) • 0x03 (Operation started)

Basic call flow for DTMF digits played during a packet call

The following call flow illustrates how the STN message is used to play DTMF digits during an active packet call. It shows a Softswitch setting up a packet call and then arbitrarily requesting that the MultiVoice Gateway at the far end play three DTMF digits: 1, 2 and 3.

```
 TAOS Unit                  TAOS Unit
 (incoming)   Softswitch    (outgoing)
 ----------   ----------    ----------
     |             |             |
     |             |    RCCP     |
     |             |------------>|
     |             |             |
     |             |    ACCP     |
     |             |<------------|
     |    RCCP     |             |
     |<------------|             |
     |             |             |
     |    ACCP     |             |
     |------------>|             |
     |             |             |
```

```
|      RTP     |              |
|<--------------------------->|voice exchange
|              |              |begins
|              |              |
|              |     STN      |
|              |------------->|Tag 0x33 = "123"
|              |              |
|              |     ASTN     |operation started
|              |<-------------|Tag 0x36 = 0x03
|              |              |
|              |              |"1" played
|              |              |
|              |              |"2" played
|              |              |
|              |              |"3" played
|              |              |
|              |     ASTN     |operation successful
|              |<-------------|Tag 0x36 = 0x00
|              |              |
```

Call flow for out-of-band DTMF transport

The following call flow illustrates how DTMF playout is used in conjunction with in-call DTMF detection to achieve true out-of-band DTMF transport using IPDC signalling.

In this example, the call originator enters two DTMF digits, 1 and 2, after the voice exchange has begun. When performing out-of-band DTMF transport, it is necessary to remove the entered DTMF from the RTP stream. To do so, set the `voip` profile on the MultiVoice Gateway as follows:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set dtmf-tone-passing = dtmf-tone-passed-outofband
admin> write
VOIP/{ 0 0 } written
```

The call flow for out-of-band DTMF transport is as follows:

```
                TAOS Unit                       TAOS Unit
               (incoming)      Softswitch      (outgoing)
               ----------      ----------      ----------
                    |              |              |
                    |              |              |
                    |              |              |
                    |              |     RCCP     |
                    |              |------------->|
                    |              |              |
                    |              |     ACCP     |
                    |              |<-------------|
DTMF detection      |     RCCP     |              |
enabled             |<-------------|              |Tag 0x75 = 0x01
                    |              |              |
                    |     ACCP     |              |
                    |------------->|              |
```

```
                |             |             |
                |     RTP     |             |
                |<--------------------------->|voice exchange
                |             |             |begins
                |             |             |
user enters     |     NTN     |             |
DTMF digit "1"  |------------>|             |Tag 0x33 = "1"
                |             |             |
                |             |     STN     |
                |             |------------>|Tag 0x33 = "1"
                |             |             |
                |             |    ASTN     |
                |             |<------------|Tag 0x36 = 0x03
                |             |             |
                |             |             |
                |             |             |"1" played
                |             |             |
                |             |    ASTN     |
                |             |<------------|Tag 0x36 = 0x00
                |             |             |
user enters     |     NTN     |             |
DTMF digit "2"  |------------>|             |Tag 0x33 = "2"
                |             |             |
                |             |     STN     |
                |             |------------>|Tag 0x33 = "2"
                |             |             |
                |             |    ASTN     |
                |             |<------------|Tag 0x36 = 0x03
                |             |             |
                |             |             |
                |             |             |"2" played
                |             |             |
                |             |    ASTN     |
                |             |<------------|Tag 0x36 = 0x00
                |             |             |
                |             |             |
```

An STN request for DTMF playout over VoIP is accepted only for an active VoIP call in voice exchange mode. It is rejected for an active VoIP call that is performing pre-call DTMF collection, playing a pre-call announcement or playing a break-in announcement.

Only the `Apply` command is allowed—`Cancel` is not allowed.

The operation is allowed regardless of the setting of the `dtmf-tone-passing` parameter in the `voip` profile (that is, `inband`, `out-of-band` or `rfc2833`).

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Error Handling

In addition to the existing conditions whereby an STN request can be rejected, the following error responses are generated:

Table 2-21. Error handling

Softswitch will receive:	For any of the following:
MRJ with Tag 0xFE (Cause Code) = 0x65 (Wrong Message For State)	• STN while a VoIP call is performing pre-call DTMF detection. • STN while a VoIP call is playing a pre-call voice announcement. • STN while a VoIP call is playing a break-in voice announcement.
ASTN with Tag 0x36 (Completion Status) = 0x01 (Operation Failed)	• STN with Tag 0x4A (Apply/Cancel Tone) = 1 (Cancel)

IPDC country-specific call-progress tone playout for VoIP

IPDC defines a specification for playing tones, such as DTMF or arbitrary frequencies towards the PSTN. TAOS provides a means for signaling one of several call progress tones (for example, dial tone, alerting, busy, network busy and unobtainable) using a MultiDSP slot card.

You can also configure the unit to generate a country-specific call progress tone. Country-specific call progress tones can be played as

- Local (to the calling party) playout of remote country-specific call progress tones, which is explained in this section.
- Carry back the call progress tones via RTP, which is already supported in the current IPDC VoIP implementation in TAOS using existing IPDC messaging.

STN message that supports country-specific call-progress tones

A tone type that has been added to the STN (Send tones or announcement) message includes a a country identifier.tone string that identifies one of the call progress tones. TAOS uses the tone string and country identifier to generate a country-specific call progress tone.

The STN message that supports generating country-specific call-progress tones is defined in IPDC specification *Level 3 Communications, Internet Protocol Device Control (IPDC), Revision 0.15.*

Tag 0x49 (tone type)

The proprietary tag value of 0x41 (Call Progress Tone) has been added to the tone type tag 0x49. The tag value minimizes conflict with any future tag values that might be added in the future by the IPDC community at large. Value 0x41 signals that the STN message contains a request to apply or cancel a call progress tone.

Associated with this value for tag 0x49 is a set of additional values used to indicate the type of call progress tone to play. These values are specified in tag 0x33 (tone string) whenever 0x49 has value 0x41, as follows:

Call progress tone	Tag 0x33 value
Dial tone	"1"
Alerting tone	"2"
Busy tone	"3"
Network busy tone	"4"
Unobtainable tone	"14"

The next available non-proprietary tag value for tag 0x49 is 0x08.

Note TAOS also provides support for two other call progress tones (that is, pin tone and error tone). However, these tones are not country-specific and are relevant only for the H.323 VoIP implementation in TAOS.

Tag 0xC1 (Country Identifier)

The proprietary tag, 0xC1, has also been added to the STN message whenever the STN message contains a request to apply or cancel a call progress tone (that is, when tag 0x49 has the value 0x41).

The following table lists the value of the 0xC1 tag generated in the STN message according to the setting of the `country` parameter in the `system` profile:

Parameter setting	Tag 0xC1 value
argentina	0x01
australia	0x02
belgium	0x03
china	0x04
costa rica	0x05
finland	0x06
france	0x07
germany	0x08
hong kong	0x09
italy	0x0A
japan	0x0B
korea	0x0C
mexico	0x0D
netherlands	0x0E
new zealand	0x0F
singapore	0x10
spain	0x11
sweden	0x12

Parameter setting	Tag 0xC1 value
switzerland	0x13
uk	0x14
us	0x15
brazil	0x16

The value of tag 0xC1 is used as the country of origin when determining the frequency, duration, and cadence of the call progress tone. If tag 0xC1 is omitted, the value of the country parameter in the system profile is used as the country of origin.

STN/ASTN usage

The semantics of the STN and ASTN (Completion result of STN command) messages remain the same as for the other tones and announcements.

- To start a call progress tone, send an STN message with tag 0x4A set to 0x00 (Apply Tone).
- To stop a call progress tone, send an STN message with tag 0x4A set to 0x01 (Cancel Tone).

All of the above five call progress tones play continuously until explicitly canceled.

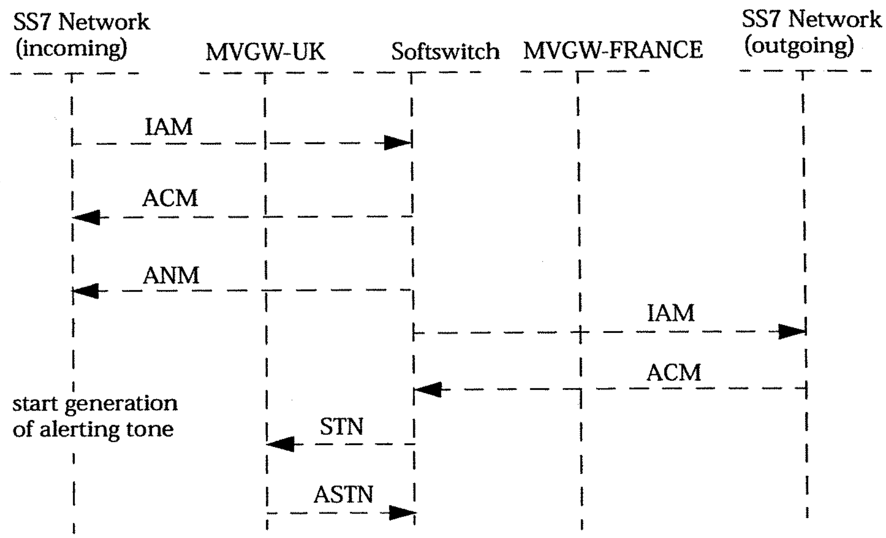
Note If the universal gateway is unable to accept an STN call progress tone request, it responds with an ASTN with status 0x01 (operation failed).

Sample Call Flows — ringing, then answered scenario

If you network consists of two universal gateways, one in the United Kingdom (MVGW-UK) and one in France (MVGW-FRANCE), with one Softswitch, and the calling party is in the UK and the called party is in France, to allow the calling party to hear an alerting tone specific to France, Softswitch sends MVGW-UK an STN with tag 0xC1 = 0x07, tag 0x49 = 0x41, tag 0x33 = "2", tag 0x4A = 0x00 and other tags as required for STN.

Softswitch received tag 0xC1 = 0x07 from MVGW-FRANCE when MVGW-FRANCE sent an NSUP to the Softswitch.

Note The NSUP message is unsupported.



Message contents:

Softswitch->MVGW: (STN - apply "alerting" call progress tone)

```
Protocol=0x4b, Correlator (4): 00000000
Message: 0x0073
Tag ID=0x07, Data (2): 00 01
Tag ID=0x0d, Data (2): 00 01
Tag ID=0x15, Data (2): 00 01
Tag ID=0x49, Data (1): 41
Tag ID=0x4a, Data (1): 00
Tag ID=0x32, Data (1): 01
Tag ID=0x33, Data (1): 32
Tag ID=0xc1, Data (1): 07
```

MVGW->Softswitch: (ASTN - command started)

```
Protocol=0x4b, Correlator (4): 00000000
Message: 0x0074
Tag ID=0x07, Data (2): 00 01
Tag ID=0x0d, Data (2): 00 01
Tag ID=0x15, Data (2): 00 01
Tag ID=0x36, Data (1): 03
```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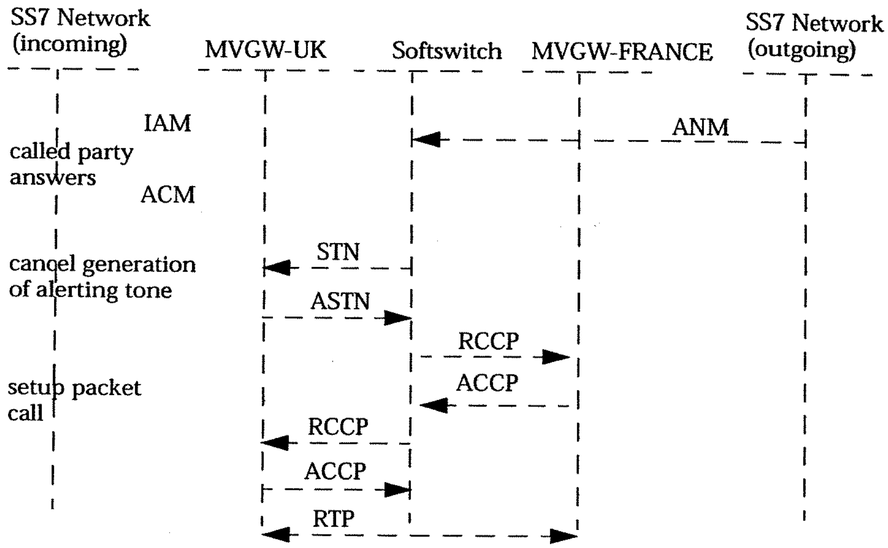
When the called party answers, Softswitch stops the alerting tone by sending MVGW-UK an STN with the same tags as above except tag 0x4A = 0x01. The packet call can then be set up.



Message contents:

Softswitch->MVGW: (STN - cancel "alerting" call progress tone)

```
Protocol=0x4b, Correlator (4): 00000000
Message: 0x0073
Tag ID=0x07, Data (2): 00 01
Tag ID=0x0d, Data (2): 00 01
Tag ID=0x15, Data (2): 00 01
Tag ID=0x49, Data (1): 41
Tag ID=0x4a, Data (1): 01
Tag ID=0x32, Data (1): 01
Tag ID=0x33, Data (1): 32
Tag ID=0xc1, Data (1): 07
```

MVGW->Softswitch: (ASTN - command completed)

```
Protocol=0x4b, Correlator (4): 00000000
Message: 0x0074
Tag ID=0x07, Data (2): 00 01
Tag ID=0x0d, Data (2): 00 01
Tag ID=0x15, Data (2): 00 01
Tag ID=0x36, Data (1): 00
```

Sample Call Flows — busy, then hangup scenario

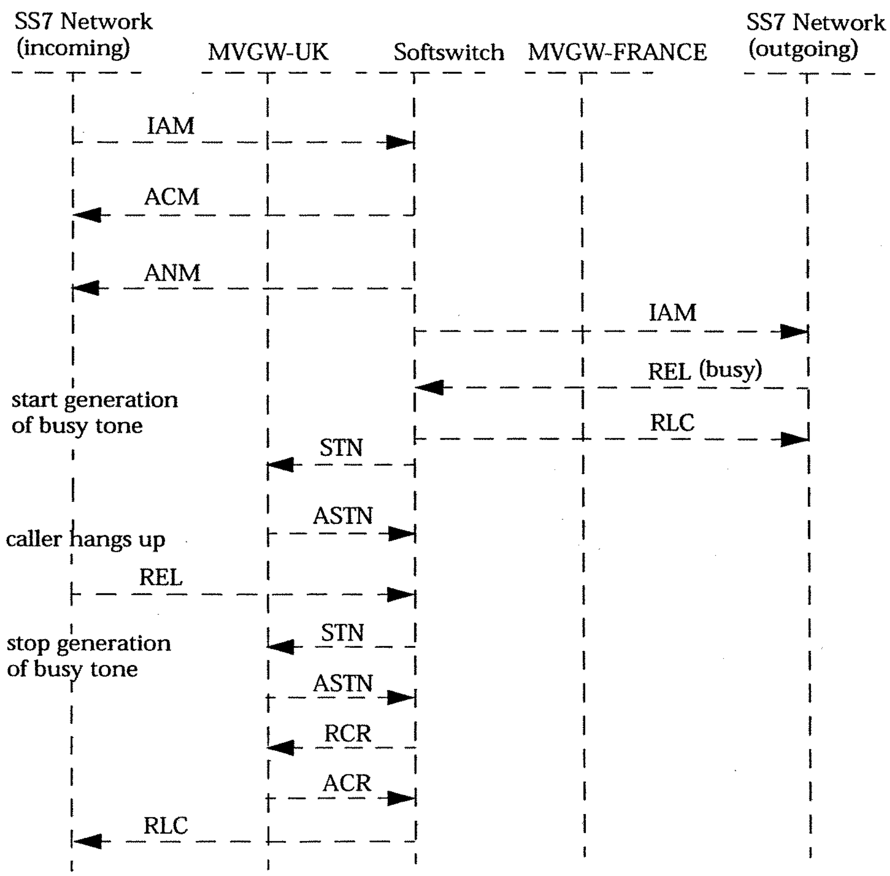
If call setup never progresses to the point where a packet call is actually setup (that is, an RCCP is sent and accepted by the universal gateway), but at least one call progress tone has been generated, and it is time to take down the call, then when VoIP call persistence is enabled on the gateway, it is necessary for the Softswitch to send the

gateway an RCR at the end of the call so that the gateway may free the resources that were used to generate the call progress tone or tones.

To see if Voip call persistence is enabled, check the setting of the `ss7voip-call-persistence` in the `voip` profile:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> list
[in VOIP/{ 0 0 }]
...
ss7voip-call-persistence = yes
...
```

In the following call flow, the called party is busy. The Softswitch directs the universal gateway to generate a busy tone specific to France. After a short while, the calling party hangs up. Since VoIP call persistence is enabled, the Softswitch must send the gateway an RCR to allow it to free the resources associated with call progress tone generation when VOIP call persistence is enabled.



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